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April 12, 1957

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SUBJECT: Contract RD-94
Task Order No. 2

[redacted] 25X1

In accordance with Article 2 of the basic contract, there are forwarded herewith two (2) copies of the Monthly Progress Report for March, 1957 on Task Order No. 2 of RD-94. The report is dated April 8, 1957. This report is UNCLASSIFIED. An additional copy is being held in [redacted] by the project engineer for the use of your personnel while at this location. 25X1

In connection with this monthly progress report, the following information is submitted:

Total expenditures to 2-28-57	$19,844
Outstanding commitments as of 2-28-57	76
Funds remaining as of 2-28-57	40,396

Very truly yours,

[redacted] 25X1

Assistant Manager
Government Contract Administration

C-1243

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Monthly Progress Report

March 1957

Task Order No. 2

Contract No. RD-94

Audio Noise Reduction Circuits

The object of this project is to develop a noise reduction circuit suitable for use in separating speech intelligence from a signal containing speech and noise when the speech intelligence is masked by the noise. The proposed method involves a principle which has been used successfully to improve the signal-to-noise ratio in music reproducing or transmission systems.[1] The system used for music contains bandpass filters which pass frequencies over a range of an octave or less. These filters are used at the input and output of a non-linear element. The output of the non-linear elements contain the fundamental, and also harmonics and subharmonics of the fundamental. However, since the pass band of the input and output bandpass filters is no greater than an octave, the harmonics and subharmonics are not transmitted by the system. The function of the non-linear element is to reject all noise signals below a given amplitude or threshold level. The threshold levels of the non-linear devices in each channel can be adjusted so that, in the absence of desired signal, the noise is rejected. When the desired signal is greater than the threshold level, the non-linear elements allow the composite signal to pass. Thus, for passages of recorded music, when the music signal is below the noise level in a given frequency channel, the channel is inoperative, and its output is eliminated from the total output. Since the contribution of this channel to the total output would have been only noise, the overall noise level is reduced. When the music

1. H.F. Olson, "Electronics," Dec. 1947.

signal in a given channel is greater than the noise, the channel conducts and allows the composite signal to pass. Thus, a channel conducts only when the desired signal is greater than the noise, and rejects when noise alone is present.

In order to apply this method of noise reduction to speech, when the wide band speech signal-to-noise ratio is very low, it is necessary to find frequency regions in which there are times when the speech amplitude is greater than the noise. Although the long time average spectrum of speech is continuous, and similar in shape to the spectrum of room noise,[2] the short time spectrum of various speech sounds contains regions of maximum energy called speech formants.[3] The assumption that this method of noise reduction could be utilized for speech was based upon the belief that it would be possible to find frequency regions in which the amplitude of the speech formants would be greater than the noise a substantial part of the time.

A study has been made to determine what bandwidths are required in order to obtain speech formant amplitudes above the noise when a wide band speech sample is just intelligible in noise. It is known that for noises with a continuous spectrum it is the noise components in the immediate frequency region of the masked tone which contribute to the masking.[4] When a very narrow band of noise is used to mask a pure tone, the masking increases as the bandwidth is increased until a certain bandwidth is reached. After this, as the bandwidth is increased, the amount of masking remains constant. This bandwidth at which the masking reaches a fixed value is termed the critical bandwidth.[5] The critical bandwidth is

2. H. Fletcher, "Speech and Hearing on Communication," Van Nostrand Co., Inc., New York, 1953 (see Figures 61 and 70).

3. Op.cit. chap. 1.

4. L.L. Beranek, "The Design of Speech Communication Systems," Proc. IRE, Vol. 35, pp. 882, Sept. 1947.

5. N.R. French and J.C. Steinberg, "Factors Governing the Intelligibility of Speech Sounds," Jour.Acoust.Soc.Amer., Vol. 19, Jan. 1947 (see Figure 7).

a function of frequency. It is different when listening with one or two ears. The critical bandwidth for two ears as a function of frequency is shown by the upper curve of Figure 1. Measurements have been made using filters which were both narrower and wider than the critical bandwidth. Both pure tones and speech mixed with continuous spectrum type noises have been studied. The results of this study show that, for the narrowest permissible bands which can be used to pass speech formants, the number of times the speech formant amplitude in a given band exceeds the noise is small. Also, in these bands, the speech amplitude is never considerably greater than the noise. Since very narrow bandwidths are required to reduce the noise below the signal, the number of bands required to cover the speech spectrum is quite large. There is no satisfactory way of evaluating the effect upon speech intelligence of small contributions from many narrow bands without building a many channeled circuit and evaluating it. From the information available from studying a few channels throughout the speech spectrum, it seems possible that some improvement in intelligibility can be effected, but this improvement may prove to be small.

In view of the fact that there is no convenient way to evaluate the contributions of a few narrow band channels to speech intelligibility, a complete multi-channel system will be developed in order to determine the effectiveness of this method of improving speech intelligibility in noise. The system under development will contain approximately 80 frequency channels in the frequency range from 700 to 3200 cps. The bandwidths of these channels will be 3 db narrower than the critical bands. The bandwidths of these channels as a function of frequency are shown by the lower curve of Figure 1. The 80 channel noise reducer circuit will be housed in a seven foot relay rack in which the 80 channels are arranged on 8 chassis with 10 channels on each chassis.

During March construction of the 80 channel noise reducer has been started in the model shop. All of the sheet metal work on the chassis have been completed and the wiring is now in progress. The low pass output filters have been completed, tested and installed on the chassis. An input-output panel and a switch panel have been designed and are under construction.

In anticipation of evaluating the complete noise reducer, a noise source has been built and measured. The noise intensity per cycle is shown in Figure 2. It is planned to evaluate the noise reduction circuit using simulated white noise, room noise, and traffic noise. A filter has been built for use in conjunction with the noise source which simulates average room noise. The noise spectrum using this filter and the spectrum of average room noise are also shown in Figure 2.

The method contemplated of evaluating the noise reduction circuit is to mix speech samples with various types of noise and then endeavor to determine the threshold of intelligibility with and without the noise reducer. Many types of speech samples and groupings have been used in articulation tests to establish quantitative values of intelligibility.[6] For example, experimenters have used nonsense syllables, monosyllabic words, phonetically balanced or P.B. lists, dissyllabic words, spondee lists, sentences, and continuous discourse. Although intelligibility could be measured quantitatively by using one of the standard articulation tests, these tests are very time consuming. It is felt that satisfactory qualitative evaluation of the circuit can be obtained by using an A-B test on continuous discourse. A few paragraphs of speech of nearly constant intensity have been recorded. These recordings will be mixed with noise and the composite signal

6. I.J. Hirsh, "The Measurement of Hearing," McGraw-Hill Book Co., Inc., New York, 1952, Chapter 5.

of speech and noise will be listened to directly, and then through the noise reduction circuit. It is believed that on a direct comparison test of this type, the effectiveness of the noise reduction circuit will be obvious.

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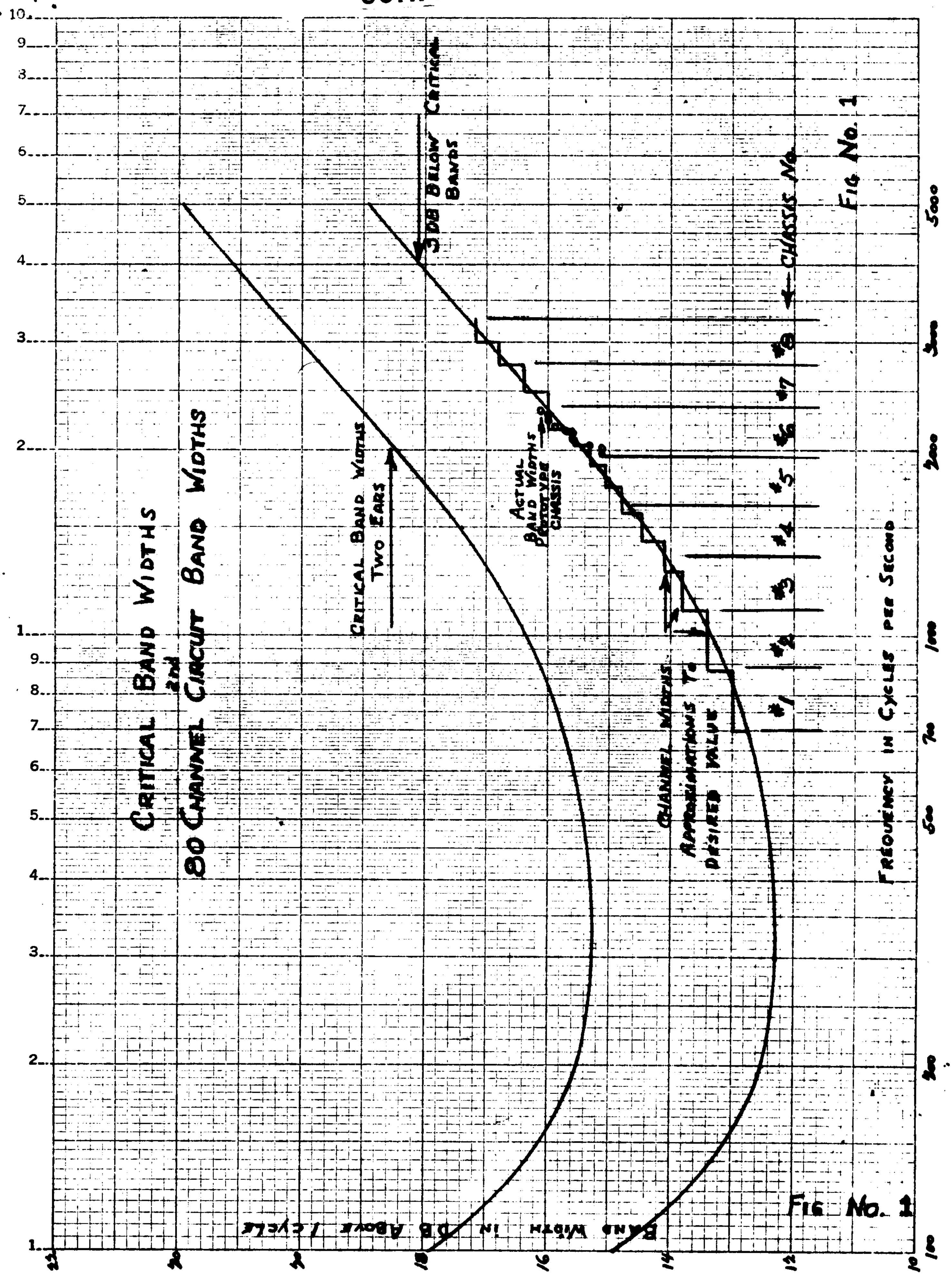
April 8, 1957



CRITICAL BAND WIDTHS
and
80 CHANNEL CIRCUIT BAND WIDTHS
3DB BELOW CRITICAL BANDS
CRITICAL BAND WIDTHS TWO EARS
ACTUAL BAND WIDTHS PROTOTYPE CHASSIS
CHANNEL WIDTHS APPROXIMATIONS TO DESIRED VALUE
← CHASSIS No.
#1 #2 #3 #4 #5 #6 #7 #8
FIG No. 1
FREQUENCY IN CYCLES PER SECOND
BAND WIDTH IN DB ABOVE 1 CYCLE
100 200 500 700 1000 2000 3000 5000
22 20 18 16 14 12 10
SEMI-LOGARITHMIC 359-61
KEUFFEL & ESSER CO. MADE IN U.S.A.
2 CYCLES X 70 DIVISIONS

FIG. No. 1

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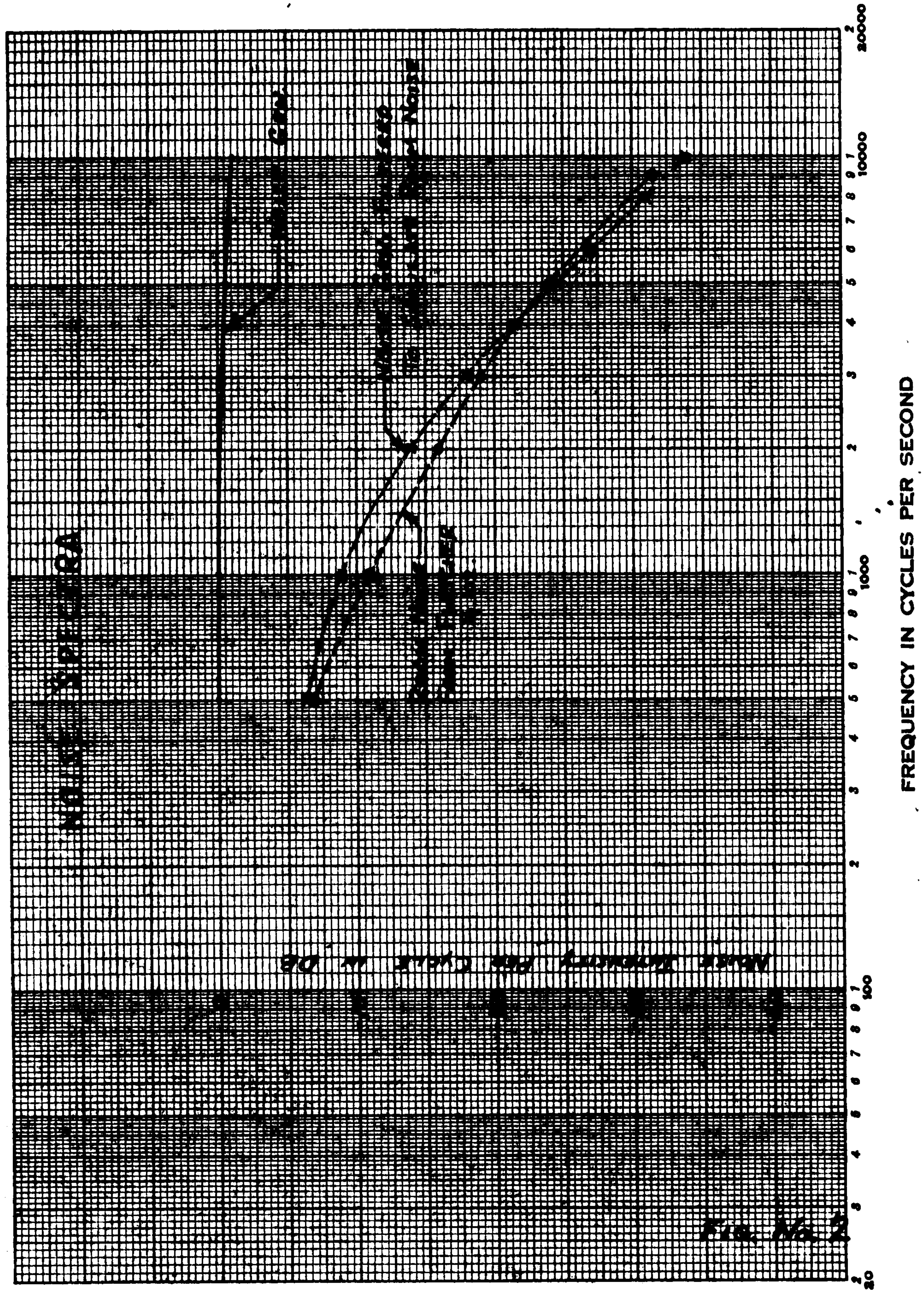

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